New Trunk and Call status reporting

Each MultiVoice Gateway reports its current call processing status as part of a Registration Request (RRQ) message to MVAM. This message includes data on trunk, trunk group and DS0 status. The initial RRQ message, sent to MVAM when a gateway is initialized, will contain a full report on all the trunks used by the physical gateway. The RRQ messages sent during keep-alive registration include only the status changes since the previous registration message.

H.323 call-specific administration messages

Call administration information is transmitted as part of the non-standard data included in registration, admission and status (RAS) messages exchanged between the gateway and gatekeeper for each call. This data consists of a set of parameters using URL encoding, as described in RFC 1738, with each parameter composed of a set of attribute value pairs.

This non standard data may include the following call administration information:

- ANI/CLID
- Conference identifier
- User PIN
- Inbound or outbound trunk identification
- Enable voice announcement playback
- Select voice announcement playback
- Internal call timer and disconnect timer settings
- Call failures
- Call results
- Trunk group and DS0 status information
- Available digital signal processors (DSPs)
- Maximum number of calls a MultiVoice Gateway may support

DS0 Status (in-service/out-of-service)

A MultiVoice Gateway reports trunk, trunk group, and DS0 information to MVAM for each trunk. This includes:

- Trunk group
- Physical address
- DS0 service status (in-service or out-of-service)

Note: A DS0 is in-service for a logical gateway when it belongs to the associated trunk group and is in the "up" state. Information regarding DS0 activity (in-use, free) is not reported via RRQ. This is handled separately. traced from the per-call trunk/DS0 reporting mentioned below.

Trunk groups and physical address (shelf, slot, etc.) information are provided to MVAM to allow dynamic tracking of DS0 activity and trunk group assignments, and provided for future support of DS0 selection by physical-address for outbound PSTN calls.

Full trunk and DS0 status reporting is performed only when necessary, enhancing gateway performance. Full RRQ's are used to report complete trunk and DS0 information, usually when a gateway is initialized or else when requested by MVAM. Lightweight RRQ's are used to

report only status changes for trunk and DS0 information. MVAM may request complete trunk and DS0 information by responding to a lightweight RRQ with a Registration Reject (RRJ) message containing a reject reason of `FullRegsitrationRequired`.

Note: Currently, trunk and DS0 status is not reported for BRI lines. Only the following information is reported for MultiVoice Gateways using BRI:

- Number of idle VOIP ports.
- value of maxCalls in VOIP profile.

Trunk and DS0 reporting (per call)

For each call processed by a MultiVoice Gateway, trunk group and physical address information for the DS0 connection are reported. This information is sent from the gateway to the gatekeeper as non-standard data in these registration, admission and status (RAS) messages, for the following call types:

Message	Call type	Trunk or DS0 information
Admission Request (ARQ)	Inbound (from PSTN)	The trunk group and physical address of the DS0 upon which the call arrived.
Bandwidth Request (BRQ)	Outbound (to PSTN)	The trunk group and physical address of the DS0 upon which the call went out.
Disengage Request (DRQ)	Inbound (from PSTN) and Outbound (to PSTN)	The physical address of the DS0. For outgoing PSTN calls, the trunk group or DS0 information may not be present.
Disengage Confirmation (DCF)	Inbound (from PSTN) and Outbound (to PSTN)	The trunk group and DS0 information for gatekeeper-initiated call terminations.

Trunk and DS0 selection (per call)

Currently, MultiVoice Gateways only support trunk-group based routing for outbound PSTN calls. To do this, using trunk groups must be enabled in the System profile of each gateway in the MultiVoice network. Each T1 must also be assigned a trunk group.

Note: Trunk groups should only be assigned at the T1 level.

The physical address information collected by the gateway for each DS0 is used currently by MVAM to dynamically track DS0 activity. It is currently not used for DS0 to DS0 linking. In the future, both trunk group and/or physical address information will be available for DS0 selection on the gateway. When this happens, trunk groups should only be used when processing both VoIP and data calls on the same gateway. Otherwise, only gatekeeper, physical-address based, DS0 routing should be used.

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Usage: This feature is enabled or disabled by assigning either `Yes`, enabling processing of call-specific administration instructions, or `No` (default), reverting global administration of VoIP calls using the values set in the Voip { X X } profile.

The following example illustrates how to enable multiple logical gateway processing on this MAX TNT:

```
admin> read voip { 0 0 }
VOIP/{ 0 0 } read

admin> set gk-mlg-control=yes

admin> write
VOIP/{ 0 0 } written
```

Dependencies: This parameter has the following dependencies:

- If `gk-mlg-control=yes`, the value of Vpn-Mode defaults to `N/A`
- If `gk-mlg-control=yes`, the value of Single-Dial-Enable defaults to `N/A`
- Changes to this parameter are effective with the next VoIP call

Location: Voip { X X }

Configuring the H.245 pipeline signal model

By tying the H.323 Alert messaging to PSTN Alerting, the gateway conveys H.245 startup information on top of the Call Proceeding message. This creates a virtual inband pipeline for call signal processing by mapping PSTN actions into H.323 actions.

In all cases, the H.245 connection information is included in all H.323 messages (Call Proceeding, Alerting, and Connect). This enables a gateway to provide the support for the following inband messaging modes:

Inband messaging mode	Description
Early alerting	In this mode, the H.323 Call Proceeding message is sent upon receipt of the Admission Confirmation (ACF) from the gatekeeper, the H.323 Alerting message is sent upon receipt of WAN inband notification from the outdialed trunk, and an H.323 Connect message is sent up receipt of the PSTN Connect message.
Slow proceeding	In this mode, the H.323 Call Proceeding message is sent upon receipt of WAN inband notification from the outdialed trunk, an H.323 Alerting message is sent upon receipt of the PSTN Alerting message, and the H.323 Connect message is sent upon receipt of the PSTN Connect message.
Fast proceeding	In this mode, which is recommended for use over high latency links, the H.323 Call Proceeding message is sent upon receipt of the Admission Confirmation (ACF) message from the gatekeeper, an H.323 Alerting message is sent upon receipt of the PSTN Alerting message, and the H.323 Connect message is sent upon receipt of the PSTN Connect message.

Usage: The Signaling-Model parameter sets the inband messaging mode used by the gateway when mapping H.323 alert messaging and PSTN alerting, and accepts the following values.

Parameter value	Description
`early-alerting`	This value (default) enables inband call signal processing on a gateway using the Early Alerting inband messaging mode.
`slow-proceeding`	This value enables inband call signal processing on a gateway using the Slow Proceeding inband messaging mode.
`fast-proceeding`	This value enables inband call signal processing on a gateway using the Fast Proceeding inband messaging mode.

The following example illustrates how to change the default value of the Signaling-Model parameter.

```
admin> read voip { 0 0 }
VOIP/{ 0 0 } read

admin> set signaling-model=fast-proceeding

admin> write
VOIP/{ 0 0 } written
```

Dependencies: Changes made to the Signaling-Model parameter take effect with the next VoIP call.

Location: Voip { x x }

Enabling fax packet redundancy

Redundant packet data is defined as the last *n* packets transmitted appended to the current packet. The value of *n* is set through the CLI using the Packet-Redundancy parameter. Once defined, this parameter controls processing of several hundred milliseconds of packet jitter and allows the optional transmission of redundant packet data for fax calls across networks experiencing instances of packet loss and packet jitter.

Assigning the Packet-Redundancy parameter a value (such as, `packet-redundancy = 4`), will cause MAX TNT to append that number of previously sent packets onto the current packet. On networks experiencing measurable packet loss, this improves the reliability of the fax transmission.

Depending upon the amount of measurable packet loss for a network, the redundancy parameter should be set accordingly:

Network condition	Recommended value(s)
Packet loss occurs in frequent bursts.	1 - 5
Occasional packet loss (less than one percent)	0 (default)
Occasional packet loss (greater than one percent)	1 - 2

The additional bandwidth required for each fax call increases proportionally to the level of redundancy, adding 50 bytes of packet data per increment. To support this feature, MultiVoice requires Real-time fax support be enabled on the MultiVoice Gateway. This may be verified by checking the Base profile for the `rt-fax-enabled=yes` entry.

This enhancement uses a slip buffer to :

- Allow MultiVoice Real-time fax to tolerate packet jitter
- Keep the modem fed with data, preventing modem underrun

Fixed sized packet format

The packet redundancy scheme uses a fixed-size packet format, consisting of a 49-byte payload, a prefixed sequence number, and a length field which precedes the payload data. When packet redundancy is enabled, *n*-length payload pairs are added at the end of the packet; where *n* is the value of the Packet-Redundancy parameter. Previously, MAX TNT sent variable length packets that were guaranteed to be zero terminated; allowing Class 1 modems to underrun gracefully.

Usage: The Packet-Redundancy parameter accepts values from 0 through 5, directing MultiVoice to append the designated number of previously transmitted fax packets to the current packet, as follows:

Parameter value	Specifies
0	No change from the default packet behavior.
1	Append and send the previous fax packet with the current fax packet.
2	Append and send the two previous fax packets with the current fax packet.
3	Append and send the three previous fax packets with the current fax packet.
4	Append and send the four previous fax packets with the current fax packet.
5	Append and send the five previous fax packets with the current fax packet.

The following example illustrates how to change the default value of the Packet-Redundancy parameter.

```
admin> read voip { 0 0 }
VOIP/{ 0 0 } read

admin> set packet-redundancy=4

admin> write
VOIP/{ 0 0 } written
```

Dependencies: The following dependencies apply to this parameter:

- Once saved, packet redundancy is enabled with the next VoIP call
- This value is set to `N/A` when `fixed-packets=no`.

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Location: Voip{x x}>Rt-Fax-Options

Enabling fixed-sized fax packets for backwards compatibility

The Fixed-Packets parameter disables use of redundant packets and the slip buffer for MultiVoice Real-time fax, enabling the pre-8.0-103 release fax packet scheme. When enabled, fax calls are processed using variable length packets that are zero terminated; allowing Class 1 modems to underrun gracefully.

The packet sequence numbering introduced in Release 8.0-103 for Real-time fax required a format change, creating high speed data packets. When these packets are absent (such as, a fax call is initiated from a MultiVoice Gateway running a pre-8.0-103 software release) the MultiVoice Gateway interprets image data as sequence data. Also the smaller packets forwarded by the new code rely on the slip buffer to keep the modem fed with data or it will drop carrier.

Usage: When the value of this parameter is `yes`, the default, the pre-8.0-103 fax packet scheme is enabled. When the value of this parameter is `no`, jitter buffering and packet redundancy for Real-time fax processing is enabled.

The following example illustrates how to enable multiple logical gateway processing on this MAX TNT:

```
admin> read voip { 0 0}
VOIP/{ 0 0} read
admin> set fixed-packets=no
admin> write
VOIP/{ 0 0 } written
```

Dependencies: The following dependencies apply to this parameter:

- Once saved, the selected packeting scheme is enabled with the next fax call
- When this value is set to `yes`, then `packet-redundancy=n/a`.

Location: voip{x x}>rt-fax-options

Configuring the fax data transmission rate

The Max-Rate parameter allows MultiVoice to modify the rate negotiation between the originating and destination fax terminals. This improves the reliability of the fax transmission by reducing the number of lost or repeated packets which occur during high rate transmissions, and reduces the required bandwidth for fax transmissions.

The fax transmission rate is regulated by modifying the content of the Digital Identification Signal (DIS) frame transmitted from the destination fax. Upon receipt of that DIS frame, the originating fax will use the data transmission rate specified by the Max-Rate parameter (or slower), and a corresponding modulation type. The content of the DIS frame is defined in the ITU Telecommunication sector standard (ITU-T) T.30, *Procedures for document facsimile transmission in general switched telephone networks.*

Changing the Max-Rate parameter modifies the high speed data transmission rate reported by the destination fax, and masks certain modulation types associated with higher fax transmission speeds. For example, when the data rate is set for 9600 bps, V.17 and V.33 are disallowed even though V.17 supports 9600 and 7200 bps. This implementation is used because:

- The DIS frame can specify only the supported modulation types for the highest selected transmission speeds at the destination fax,
- The calling fax terminal requires "training" to match the supported modulation.

Usage: Values assigned to the Max-Rate parameter cause MultiVoice to do the following:

Parameter value	Specifies
14400	Default. Mask the fax capabilities in the DIS frame that support fax data transmission at rates higher than 14,400 bps.
9600	Mask the fax capabilities in the DIS frame that support fax data transmission at rates higher than 9,600 bps.
4800	Mask the fax capabilities in the DIS frame that support fax data transmission at rates higher than 4,800 bps.
2400	Mask the fax capabilities in the DIS frame that support fax data transmission at rates higher than 2,400 bps.

The following example illustrates how to set the fax data transmission rates:

```
admin> read voip { 0 0 }
VOIP/{ 0 0 } read
admin> list rt-fax-options
[in VOIP/{ 0 0 }:rt-fax-options]
admin> set max-rate=9600
admin> write
VOIP/{ 0 0 } written
```

Dependencies: This parameter has the following dependencies:

- This parameter is N/A when `rt-fax-enable=no`.
- Changes made to this parameter are enabled for the next VoIP call.

Location: Voip{X X}->Rt-Fax-Options

Modified E1 profile settings in MAX TNT TAOS 8.0-103

The following parameters (shown with new values) are new or modified in MAX TNT TAOS 8.0-103:

```
[in E1/{ 1 1 5 }:line-interface]
signaling-mode = dtmf-r2-signaling
number-complete = 15-digits
```


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Parameter	Specifies
Signaling-Mode	This parameter has been enhanced to allow processing of Dual Tone Multi-Frequency (DTMF) tones over R2 signaling trunks by MultiVoice Gateways. This modification allows the MAX TNT to recognize and respond to either country specific R2 signaling (MFC-R2) or DTMF signaling over trunks supporting standard R2 signaling.
Number-Complete	This parameter has been enhanced to allow collection of up to 15 digits for R2 dial strings without waiting for end-of-pulse signaling.

Enabling DTMF-R2 signal processing

A new option added to the Signaling-Mode parameter allows MultiVoice Gateways to support DTMF R2 signaling generated by smaller European network switches and PBXs. MultiVoice implements DTMF tone processing using the R2 signaling standard defined by the International Telecommunications Union Telecommunication sector standard (ITU-T) Q.400, *Specifications of Signaling System R2 Definition and Function of Signals -- Forward Line Signals*.

To support DTMF-R2 detection, MultiVoice requires the following:

- Connection to E1 trunks attached to a switch that supports the ITU-T R2 signaling standard
- The switch must generate and/or relay the high-frequency/low-frequency tone combinations generated by normal touch tone dialing to the MultiVoice Gateway
- E1/R2 signaling must be enabled on the MultiVoice Gateway. This may be verified by checking the Base profile for the `r2-signaling-enabled=yes` entry

Detection of DTMF R2 signals is enabled from the E1 line profile.

DTMF tone detection

When processing tones for DTMF R2 signaling, the MultiVoice Gateway will:

- Upon detection of an inbound call, allocate a DSP for detecting DTMF tones; capturing DTMF digits as they are received from the switch.
- Upon receipt of an outbound call (from the packet network) allocate a DSP for generating DTMF tones; sending the first DTMF tone for 70ms, followed by 70ms of silence. This tone/silence sequence is repeated until all digits are sent to the telephone switch.

Usage: Setting the value of the Signaling-Mode parameter to `dtmf-r2-signaling` value enables the MAX TNT to recognize and respond to the DTMF R2 signal set during voice and data calls.

The following example illustrates how to enable multiple logical gateway processing on this MAX TNT:

```
admin> read e1 { 1 1 7 }
E1/{ 1 1 7 } read

admin> set signaling-mode=dtmf-r2-signaling

admin> write
E1/{ 1 1 7 } written
```

Dependencies: The following dependencies apply when `signaling-mode=dtmf-r2-signaling`:

- Once selected, DTMF R2 detection is enabled with the next VoIP call
- DTMF R2 detection is only supported when R2 signal processing is enabled for this MultiVoice Gateway.

Location: E1 { x x x }>Line-Interface

Collecting 15-digit dial strings

The Number-Complete parameter may now be used to configure a MAX TNT to collect 15-digit dial strings off of E1 trunks supporting inband CMF R2. This allows a MAX TNT to interoperate with European telephone systems that use E.164 addresses which are up to 15 digits long, without waiting for an end-of-pulse signal.

Previously, MultiVoice Gateways could be configured to collect dial strings of up to only 11 digits. For European networks using dial strings that were 12 digits or longer, a MultiVoice Gateway could only be configured to wait for the end-of-pulse signal to confirm it received all the dialed digits.

Usage: This parameter now accepts values from `0-digits` through `15-digits`, or `end-of-pulse` as valid entries.

The following example illustrates how to enable multiple logical gateway processing on this MAX TNT:

```
admin> read e1 { 1 1 7 }
E1/{ 1 1 7 } read
admin> set number-complete=15-digits
admin> write
E1/{ 1 1 7 } written
```

Dependencies: The following dependencies apply to this parameter:

- Changes are applied with the next VoIP call
- This parameter defaults to `N/A` when the Signaling-Mode parameter is assigned the following values:
  - `e1-kuwait-signaling`
  - `isdn`
  - `p7`
  - `dpnss`
  - `none`

Location: E1 { x x x }>Line-Interface

New VoIP profile settings in MAX TNT TAOS 8.0.1

The following parameters (shown with default values) are new or modified in MAX TNT TAOS 8.0.1:

```
[in VOIP/{ 0 0 }]
voice-ann-dir = /current
```

```
allow-g711-fallback = yes
allow-coder-fallback = yes
choose-dsp-via = voip-centric
trunk-quiesce-enable = no
early-ringback-enable = no
trunk-prefix-enable = no
```

Parameter	Specifies
Voice-Ann-Dir	Location of voice announcement files on a PCMCIA flash memory card in the MAX TNT unit. In previous releases, the value was read-only. In MAX TNT TAOS 8.0-103, administrators can create directories on the flash memory file system and specify a location for voice announcement files. See *Storing voice announcements in the FAT-16 flash memory file system* on page 36.
Allow-G711-Fallback	Enable/disable selection of the G.711 codec if the Gateway is unable to select its preferred codec. This parameter does not apply if Allow-Coder-Fallback is set to no. For details, see *Allowing fallback to alternate codecs* on page 37.
Allow-Coder-Fallback	Enable/disable selection of an alternate codec if the Gateway is unable to select its preferred codec. For details, see *Allowing fallback to alternate codecs* on page 37.
Choose-DSP-Via	*Not currently supported.*
Trunk-Quiesce-Enable	Enable/disable deactivation of a T1 PRI line when a Gateway is unavailable. For details, see *Deactivating trunks used for VoIP calls* on page 37.
Early-Ringback-Enable	Enable/disable generation of an early ringback tone on networks experiencing long call setup times. If the parameter is set to yes, the near-end Gateway plays a ringback tone to the caller as soon as a call connection is established with the far-end Gateway.
Trunk-Prefix-Enable	Enable/disable identification of the entry (ingress) trunk number to the exit (egress) Gateway or call signaling entity by prepending the ingress trunk number to the DNIS number.

Storing voice announcements in the FAT-16 flash memory file system

By default, MultiVoice callers are notified of call progress by DTMF-based tones. The tones report easily recognized call states such as ringback, busy signal, and so forth, as well as tones specific to MultiVoice, such as PIN prompt, which are not as easily recognized by callers. In previous MultiVoice releases, the MAX TNT introduced support for the playback of custom voice announcements to callers to indicate call progress. For details about how voice announcements work, and for information about managing them in the MAX TNT, see the *MultiVoice for the MAX TNT Configuration Guide* at http://www.ascend.com/doclibrary.

With MAX TNT TAOS 8.0-103, you can create directories on the flash memory file system and specify a location for voice announcement files. After creating the directory on a flash card and moving voice announcement files into it, specify the pathname in the Voice-Ann-Dir setting. For example, the following commands create a directory named messages and a subdirectory named announce on the flash card in slot 1:

```
admin> mkdir 1/messages

admin> mkdir 1/messages/announce
```

The following command loads a voice-announcement file named `busy.au` from a TFTP server at 10.10.10.10 to the `/current` directory on flash card 1 (flash card 1 is the default):

```
admin> load file network 10.10.10.10 busy.au
```

The following command moves the `busy.au` file to the new subdirectory on flash card 1:

```
admin> mv 1/current/busy.au 1/messages/announce/busy.au
```

The following commands inform the MultiVoice subsystem of the location of the voice announcement files:

```
admin> read voip { 0 0 }
VOIP/{ 0 0 } read

admin> set voice-ann-dir = /messages/announce

admin> write
VOIP/{ 0 0 } written
```

You can specify a pathname up to 40 characters long. When the system receives a request to play an announcement, it looks in the specified directory on the flash card in slot 1. If the card is not present or the voice announcement file is not found, the system looks for the specified directory on flash card 2.

Allowing fallback to alternate codecs

Voice is transmitted across an IP network as compressed audio frames. The Packet-Audio-Mode parameter in the default VoIP profile specifies the preferred audio codec used by the Gateways to compress and uncompress analog speech and digital audio frames.

In MAX TNT TAOS 8.0-103, you can set the following parameters (shown with default values) to specify how the system behaves when the preferred codec is not supported:

```
[in VOIP/{ 0 0 }]
allow-g711-fallback = yes
allow-coder-fallback = yes
```

Normally, an H.323 stack advertises a list of supported audio codecs. If the preferred codec is present on a list received from a far-end Gateway, that codec is always selected. Otherwise, the system selects an alternate codec that matches one from its supported list.

The Allow-Coder-Fallback parameter can be set to `no` to override the default system behavior and force the Gateway to reject the call if it is unable to select its preferred codec. If this parameter is set to `no`, the Allow-G711-Fallback parameter has no effect.

If Allow-Coder-Fallback parameter is set to `yes`, you can set the Allow-G711-Fallback parameter to `no` to prevent the system from selecting the G.711 codec when selecting an alternate codec. In this case, the system terminates the call if G.711 is the only available choice and it is not the preferred code. This setting affects VoIP, fax, and transparent modem calls.

Deactivating trunks used for VoIP calls

The trunk deactivation feature enables MultiVoice Gateways to automatically deactivate trunks used for VoIP calls when a Gateway becomes unavailable. This feature allows Gatekeepers in the MultiVoice network to route calls to other available Gateways, to use network resources more efficiently and improve service quality for users.

Note: In this release, only T1 trunks that use ISDN PRI signaling and have been configured for VoIP can be deactivated system-wide by using this feature.

Trunk deactivation prevents the PSTN switch from routing subsequent calls to the trunks configured for VoIP. Current calls remain active until those calls are terminated by the caller or PSTN. When trunk deactivation is enabled, trunks configured to accept VoIP calls are made unavailable to the PSTN under the following conditions:

- A Gateway cannot register with either a primary or secondary Gatekeeper.
- A Gateway's trunk connection with the PSTN is unavailable, so that Gateway is forced to unregister itself from its Gatekeepers.

Previously, when a Gateway could not register with the primary and secondary Gatekeeper, the caller heard a fast busy signal because the PSTN switch continued to route calls to the trunks on that Gateway. Deactivating the trunk changes the trunk state to inform the PSTN switch aware that those trunks are not available.

Previously, when a VoIP call could not connect because a trunk was not operating, the caller heard a fast busy signal, because the Gatekeeper continued to route calls to that Gateway as long as it remained registered. Deactivating the trunk forces the Gateway to unregister from all known Gatekeepers, which causes the Gatekeepers to reroute new calls to other Gateways. When any one of the Gateway's trunks comes back in service, that Gateway starts registering itself with one of its known Gatekeepers. The Gatekeeper then begins to route calls to this Gateway.

The following commands enable trunk deactivation for T1 PRI lines configured for VoIP:

```
admin> read voip { 0 0 }
VOIP/{ 0 0 } read

admin> set trunk-quiesce-enable = yes

admin> write
VOIP/{ 0 0 } written
```

Enabling early ringback

For certain VoIP network configurations, such as satellite IP networks, wireless networks, or networks using channel-associated signaling (CAS) trunks, call setup times can be quite long. Callers might hang up before the call completes because they hear no call progress tones until RTP carries ringback from the far end PSTN. Early ringback allows the MAX TNT to generate a ringback tone locally, as soon as the call is started on the far-end Gateway.

Note: Early ringback is intended for use only on networks that experience long call setup times. Its use for other network configurations is not recommended, and might result in erroneous ring-to-busy and ring-to-failure announcements.

The following commands enable early ringback:

```
admin> read voip { 0 0 }
VOIP/{ 0 0 } read

admin> set early-ringback-enable = yes

admin> write
VOIP/{ 0 0 } written
```

Trunk prefixing

Trunk prefixing enables the MAX TNT to identify the entry (ingress) trunk number to the exit (egress) gateway or call signaling entity by prepending the ingress trunk number to the DNIS number. Trunk groups must be in use system-wide.

When trunk prefixing is enabled, the system obtains the trunk group number of the ingress T1 trunk from the `trunk-group` setting in the T1 line profile, and prepends it to the detected DNIS number. The Q.931 Called Party Number information element (IE) in an H.225/Q.931 SETUP message then contains the DNIS number prefixed by the incoming trunk number. The destination address value of the SETUP user-to-user information element (UUIE) is not currently encoded.

For example, the following commands enable trunk prefixing, beginning with the next VoIP call the MAX TNT receives:

```
admin> read voip { 0 0 }
VOIP/{ 0 0 } read

admin> set trunk-prefix-enable = yes

admin> write
VOIP/{ 0 0 } written
```

Real-time fax

MultiVoice real-time fax uses the VoIP framework for call establishment, fax initiation, and detection of an incoming fax call.

Note: Real-time fax communications require guaranteed quality of service between the two fax-capable Gateways. The packet loss on the network must be less than 1%.

Real-time fax calls begin when a VoIP call is placed from an originating fax machine to the answering machine. If the MAX TNT is configured to perform out-of-band dual tone multifrequency (DTMF) signaling, a DSP automatically enables inband DTMF signaling at the start of the fax call. When the destination fax machine picks up the call and sends an answer tone, known as a CED tone, the destination Gateway detects this tone and initiates a switchover to real-time fax on both itself and the Gateway at the other end of the call. When the switchover is complete, the fax transmission proceeds normally.

You must create the appropriate coverage areas on the MultiVoice Access Manager to ensure that fax calls are routed between Gateways that are fax capable. For details, see the *MultiVoice Access Manager User's Guide* at http://www.ascend.com/doclibrary.

Overview of real-time fax settings

Following are the parameters (shown with default values) for enabling and improving the performance of real-time fax processing. Changes to these parameters take effect with the next VoIP call.

```
[in VOIP/{ 0 0 }:rt-fax-options]
rt-fax-enable = no
ecm-enable = yes
low-latency-mode = yes
command-spoof = yes
local-retransmit-lsf = yes
```

Parameter	Specifies
RT-Fax-Enable	Enable/disable Real-time fax call processing. When the parameter is set to `no` (the default), fax tones are passed as if they were normal voice samples, and the other parameters in the subprofile are not applicable. When the parameter value is set to `yes`, this MAX TNT switches over from voice session to fax upon detection of a CED tone or V.21 HDLC flag.
ECM-Enable	Enable/disable error correction mode (ECM) for real-time fax calls. When the parameter is set to `yes` (the default), fax frames can be retransmitted in the event that a frame is not received correctly. ECM frames are relayed end to end between terminals. Setting the parameter to `no` disables ECM, so fax frames containing errors are not corrected.
Low-Latency-Mode	Enable/disable low latency mode for real-time fax operations over networks with low packet loss and low latency characteristics. Low latency mode allows operation on networks with less than 2.5 seconds or less of aggregate latency between pages. When the parameter is set to `no`, a minimum of 10 seconds delay is added to processing fax calls to allow interpretation of T.30 frames and implement spoofing.
Command-Spoof	Enable/disable spoofing of certain fax commands. Command spoofing is a method of improving performance and reducing fax errors on low latency networks.
Local-Retransmit-LSF	Enable/disable local retransmission of a low speed fax frame if no response is detected from the destination fax. This is designed to reduce fax transmission errors on low packet loss networks

In an SS7 environment, values in IPDC messages override corresponding call management settings in the default VoIP profile.

Example real-time fax configuration

For example, the following commands enable Real-time fax call processing and leave all performance parameters enabled:

```
admin> read voip { 0 0 }
VOIP/{ 0 0 } read

admin> set rt-fax-options rt-fax-enable = yes

admin> write
VOIP/{ 0 0 } written
```

Transparent modem

MultiVoice supports a transparent data mode that enables users to run a modem on a VoIP channel, regardless of the audio codec that is in use.

Overview of transparent modem settings

Following is the parameter for enabling the transparent modem features, shown with the default setting:

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```
[in VOIP { 0 0 }]
g711-transparent-data = no
```

Parameter	Specifies
G711-Transparent-Data	Enable/disable transparent modem mode. When the parameter is set to `yes`, when the MAX TNT detects a modem in a VoIP channel, the unit transparently requests end-to-end G.711 encoding and bandwidth for the call, in a process similar to that used by real-time fax . The echo cancelers are disabled when the MAX TNT enters this mode, thus providing transparent G.711 encoding. The data is encoded transparently as an audio-mode type, either G.711 µ-law (64Kbps) or G.711 A-law (64Kbps). Settings take effect with the next incoming PSTN call. A separate license is not required for this feature.

In an SS7 environment, values in IPDC messages override corresponding call management settings in the default VoIP profile. For information about IPDC support for transparent modem, see *IPDC message support for fax and transparent modem* on page 43.

Example transparent modem configuration

The following commands enable the transparent modem feature on VoIP channels:

```
admin> read voip { 0 0 }
VOIP/{ 0 0 } read

admin> set g711-transparent-data = yes

admin> write
VOIP/{ 0 0 } written
```

Using transparent modem with real-time fax

If the MAX TNT has been licensed for real-time fax, users can run either a high-speed modem with speeds greater than 2400 bps or a fax terminal in the VoIP channel. This capability provides a fallback for real-time fax transmissions. Both fax terminals and high-speed modems transmit a single tone when they answer a call, but each type of equipment uses a different tone. The MAX TNT detects the type of equipment in use on the basis of its answer tone. When it detects the equipment answering the call, the MAX TNT sends H.245 request-mode messages to request a switchover from the current audio codec to either G.711 with no echo canceler (for transparent modem) or fax data mode (for real-time fax).

Transparent data is encoded as an audio-mode type, either G.711µ-law (64Kbps) or G.711 A-law (64Kbps). Real-time fax (if supported) is encoded as a fax data-mode type.

Note: Transparent data mode introduces an H.245 request-mode message that is not backward compatible with the real-time fax feature provided by previous MultiVoice releases. To interoperate with a Gateway using transparent mode, all Gateways must be upgraded to MAX TNT TAOS 8.0-103.

Example real-time fax and transparent modem configuration

The following commands enable both real-time fax and the transparent modem feature for high-speed modems:

```
admin> read voip { 0 0 }
VOIP/{ 0 0 } read

admin> set rt-fax-options rt-fax-enable = yes

admin> set g711-transparent-data = yes

admin> write
VOIP/{ 0 0 } written
```

Limitation for low-speed modems

Real-time fax cannot be used concurrently with low-speed modems (2400bps or less) because these modems use the same answer tone as fax terminals. If a low-speed modem is used on a VoIP channel that is enabled for real-time fax, the Gateway detects a fax answer tone and requests T.38 encoding. The ingress Gateway (typically the Gateway on which the modem call originated) can accept the T.38 encoding request or reject the request, which causes the egress Gateway to terminate the call.

IPDC message support for modifying parameters

With MAX TNT TAOS 8.0-103, MAX TNT units provide limited support for IPDC messages used to modify the following values for VoIP calls. The request modify packet pass-through call (RMCP) message (`0x0015`) and accept modify packet pass-through call (AMCP) message (`0x0016`) allow modification of the following values for VoIP calls.

- VoIP encoding type (G.711μ-law, G.711A-law G.729, or G.723).
- Packet loading rate in frames per packet (value depends on VoIP encoding type)
- Source port type (currently, only the `SCN` value is supported).
- Destination port type (currently, only the `RTP` value is supported).
- Listen IP address.
- Listen RTP port number.
- Send IP address.
- Send RTP port number.

The MAX TNT can allocate its own system IP address as the listen IP address and RTP port and can specify its own send address and RTP port. For VoIP calls, you must avoid routing RTP traffic through the MAX TNT shelf controller. For that reason, when allowing the MAX TNT Gateway to allocate its own address, you must set the System-IP-Addr parameter in the IP-Global profile to an interface address other than the shelf-controller Ethernet port. For example, the following commands set the system address to the address of a port on an Ethernet card in slot 12:

```
admin> get ip-interface { { 1 12 1 } 0} ip-address
[in IP-INTERFACE/{ { shelf-1 slot-12 1 } 0 }:ip-address]
ip-address = 1.1.1.1/24

admin> read ip-global
IP-GLOBAL read

admin> set system-ip-addr = 1.1.1.1/24

admin> write
IP-GLOBAL written
```

In addition, you must make sure that VoIP calls can always find a route to the next-hop Gateway on the path to the destination VoIP Gateway. The route can be learned dynamically or configured as a static route. Many sites choose to configure default routes for VoIP traffic, so that RTP packets are never dropped due to lack of routing information. For example, the following commands configure a default route named VoIP to a next-hop Gateway at 2.2.2.2:

```
admin> new ip-route voip
IP-ROUTE/voip read

admin> set gateway = 2.2.2.2/24

admin> write
IP-ROUTE/VoIP written
```

IPDC message support for fax and transparent modem

Previously, transparent data for fax and modem calls was available only in an H.323 environment or for IPDC calls running G.711 codecs for VoIP. In this release, IPDC message request packet pass-through call (RCCP), accept packet pass-through call (ACCP), request modify for packet pass-through call (RMCP), and accept modify packet pass-through call (AMCP) messages enable an SS7 signaling gateway to direct the MAX TNT to enter T.38 fax mode or transparent modem mode on the basis of tone detection. In addition, the signaling gateway can control echo cancelation by disabling it or setting it to 32 milliseconds on a per-call basis.

The notify tone (NTN) message is used to notify the signaling gateway when an asynchronous fax or modem tone is detected. The MAX TNT sends this message to the signaling gateway if either fax or modem tone detection is enabled and the unit sees the tone. The MAX TNT detects fax tone if `rt-fax-enable` is set to `yes` in the default VoIP profile or if it receives the relevant IPDC message from the signaling gateway.

The MAX TNT detects modem tone if `g711-transparent-data` is set to `yes` in the default VoIP profile or if it receives the relevant IPDC message from the signaling gateway.

For an introduction to the real-time fax feature, see “Real-time fax,” on page 39. For an introduction to the transparent modem feature, see “Transparent modem,” on page 40.

New trunk features for VoIP calls

With MAX TNT TAOS 8.0.1, MAX TNT units provide a configurable timer for T1 lines that use inband signaling, a true connect feature to avoid charges for VoIP calls, and a calling line ID (CLID) generated by the MultiVoice Access Manager (MVAM).

Configurable interdigit timer for T1 inband signaling

When a T1 line uses inband signaling, you can enable Collect-Incoming-Digits to cause the DSP to decode the calling and called DTMF digits on the line, making DNIS and CLID information available for authentication and accounting. Following is the relevant parameter, shown with a sample setting:

```
[in T1/{ any-shelf any-slot 0 }:line-interface]
collect-incoming-digits = yes
```

In previous releases, when this feature was enabled, the T1 DSP always waited for 3 seconds after collecting the last digit before considering DNIS or automatic number identification

(ANI) collection complete. This 3-second timeout slowed down call setup times, and was unnecessary when a switch or PBX was generating the DTMF DNIS/ANI information with digit and interdigit times much smaller than 3 seconds. To improve call setup times, especially for VoIP calls with single-stage-dial, you can now configure the timeout for collecting incoming digits. Following is the relevant parameter, shown with its default value:

```
[in T1/{ any-shelf any-slot 0 }:line-interface]
t1-inter-digit-timeout = 3000
```

Parameter	Specifies
T1-Inter-Digit-Timeout	Number of milliseconds the T1 DSP waits between digits before considering DNIS/ANI collection complete. For backward compatibility, the default is 3 seconds. The valid range is 100 to 6000 milliseconds. The setting takes effect with the next incoming call. Specifying a lower value improves call setup times. This is especially important for VoIP calls with single-stage-dial. This parameter does not apply unless Collect-Incoming-Digits is set to `yes`.

For example, the following commands specify a timeout of half a second:

```
admin> read t1 { 1 2 3 }
T1/{ shelf-1 slot-2 3 } read

admin> set line-interface collect-incoming-digits = yes

admin> set line-interface t1-inter-digit-timeout = 500

admin> write
T1/{ shelf-1 slot-2 3 } written
```

Delaying charges until call is answered (true connect)

In earlier releases, incoming VoIP calls from the PSTN were connected at the near end Gateway before any H.323 signaling was sent to the far end Gateway. As a result, a PSTN charge was incurred at the time of connection to the near-end Gateway, before the called party received and answered the call from the far-end Gateway.

Now, you can change this behavior by enabling true connect. When this feature is enabled, alerting and connect messages sent to the PSTN switch are delayed to match the equivalent H.323 signaling to avoid incurring charges before a VoIP call has been answered.

The true connect feature requires a default call type of VoIP on T1 or E1 trunks accepting incoming VoIP calls. Following are the relevant parameters, shown with sample settings:

```
[in VOIP { 0 0 }]
true-connect-enable = yes

[in T1/{ shelf-1 slot-10 1 }:line-interface]
default-call-type = voip

[in E1/{ shelf-1 slot-11 1 }:line-interface]
default-call-type = voip
```

Parameter	Specifies
True-Connect-Enable	Enable/disable delay of PSTN alerting and connect messages to match the equivalent H.323 alerting and connect messages. The default setting is `no`, which results in the caller incurring a PSTN charge at the time of connection to the near-end Gateway, before the called party has received and answered the call from the far end Gateway. If set to `yes`, an alerting message is sent to the ingress PSTN switch only when an H.323 alerting message is received on the ingress VoIP Gateway. Similarly, a PSTN connect message is sent only when the H.323 VoIP call has been answered. This ensures that no charges are incurred for incomplete calls. The setting takes effect with the next incoming call. It has no effect on outbound calls.
Default-Call-Type	Must be set to VoIP for T1 or E1 trunks with incoming VoIP calls that require true connect. Note that setting this parameter to VoIP causes *all* calls received on the trunk to be mapped to VoIP.

For example, the following commands enable delayed PSTN alerting and connect messages on trunk lines configured with a default VoIP call type:

```
admin> read voip { 0 0 }
VoIP { 0 0 } read

admin> set true-connect-enable = yes

admin> write
VoIP { 0 0 } written
```

Note: For ISDN trunks, Lucent recommends that you set the T310 timer on the telephone company switch or PBX to 30 seconds or greater when using the true connect feature. because the T310 timeout value includes the time that the called party's telephone is ringing, a 10-second timeout can cause the near-end Gateway to disconnect the call too soon.

When the true connect feature is enabled and a VoIP call fails before the PSTN call is fully connected, the Gateway is still able to send an appropriate tone or voice announcement to the caller.

Gatekeeper CLID substitution

When MultiVoice Gateways are connecting VoIP calls, they can transmit a calling line ID (CLID) generated by the MVAM software on the Gatekeeper instead of the PSTN-generated CLID collected on the trunk line. CLID substitution allows the MultiVoice network to provide the appropriate E.164 address for both the called and calling telephone numbers to the respective PSTN, and for use by external applications.

In certain configurations in which the Gateways connecting the call reside in different area codes or countries, the CLID received from the PSTN must be changed to provide the appropriate calling number information to the local carrier, or to call management and billing applications.

When the MVAM receives the CLID from a Gateway, it translates the CLID to the appropriate dial string, adding or removing country codes and area codes as appropriate for the respective locations of the callers. The Gatekeeper then reports the revised CLID to the Gateways as part of the admission confirmed (ACF) message.

RT-24 (proprietary) codec support

The RT-24 codec is a Lucent Technologies proprietary audio codec that compresses speech samples from 64Kbps pulse code modulation (PCM) to 2.4Kbps, reducing the effective bandwidth required for transmission across the IP network.

This codec uses a 22.5-millisecond audio frame, and encapsulates audio at 8 bytes per frame. The decoder produces 180 samples of audio from the 8-byte encoder output. The RT-24 codec is available for both H.323 VoIP calls and SS7 VoIP calls.

When the RT-24 codec is selected, the MultiVoice Gateway attempts to determine if that codec is supported by the other Gateway during H.245 capability negotiation. If both sides agree to use RT-24 as the preferred codec, both Gateways enable RT-24 on the allocated DSPs to compress and decompress audio after the H.245 open logical channel message is exchanged.

Note: RT-24 is a Lucent Technologies proprietary codec, which is available only on MultiVoice Gateways running MAX TNT TAOS 8.0-103. MultiVoice cannot use this codec when communicating with a third-party VoIP gateway.

To enable RT-24 audio processing, set the packet-audio-mode parameter in the default VoIP profile to the selected codec as illustrated by the following:

```
admin> read voip { 0 0 }
VOIP/{ 0 0 } read

admin> set packet-audio-mode = rt24

admin> write
VOIP/{ 0 0 } written
```

G.728 codec support

G.728 is a Low-Delay Code Excited Linear Prediction (LD-CELP) based audio codec that provides toll-quality audio at a bit-rate of 16Kbps. With a frame size of only 2.5 milliseconds, G.728 also has a very low delay. Although the MultiVoice implementation of G.728 uses a frame size of 5 milliseconds, the bitstream from the audio codec is the same as described in the ITU-T standard and can thus be decoded by any G.728 decoder.

Each MultiDSP card supports a maximum of 48 simultaneous G.728 calls for both H.323 VoIP and SS7 VoIP call processing.

When the G.728 codec is selected, the MultiVoice Gateway attempts to determine if the G.728 codec is supported by the other Gateway during H.245 capability negotiation. If both sides agree to use G.728 as the preferred codec, both Gateways use G.728 to compress and decompress audio after the H.245 open logical channel message is exchanged.

Note: Although MultiVoice uses a 5-millisecond frame for G.728 processing, it is compatible with any third-party G.728 decoder. However, if a MultiVoice Gateway attempts to communicate with a third-party VoIP gateway transmitting an odd number of 2.5 millisecond frames per IP packet, the call will fail.

When you enable G.728 audio processing in this release the Silence-Det-Cng parameter must be set to no (its default value). The following commands enable G.728 processing:

```
admin> read voip { 0 0 }
VOIP/{ 0 0 } read

admin> set packet-audio-mode = g728
```


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```
admin> set silence-det-cng = no

admin> write
VOIP/{ 0 0 } written
```

SNMP: Support for the VoIP MIB (ascend 28)

The VoIP MIB enables network management stations to monitor MultiVoice Gateway operations using SNMP. Attributes in the MIB can be obtained by SNMP Get and Get-Next operations. The MIB uses the following object identifiers for identifying MultiVoice Gateway or Gatekeepers to a network manager:

- voipCfgGroup (voipGroup 1)
- voipCfgGkGroup (voipCfgGroup 1)
- voipCfgGwGroup (voipCfgGroup 2)

The MIB uses the following tables for identifying MultiVoice Gatekeeper and Gateway functions.

```
voipCfgGkTable OBJECT-TYPE (voipCfgGkGroup 1)
   SYNTAX SEQUENCE OF VoipCfgGkEntry
   ACCESS not-accessible
   STATUS mandatory
   DESCRIPTION A list of entries for H323 network Gatekeeper.

voipCfgGkEntry OBJECT-TYPE (voipCfgGkTable 1)
   SYNTAX VoipCfgGkEntry
   ACCESS not-accessible
   STATUS mandatory
   DESCRIPTION An entry holding information about the Gatekeeper for
   the system.
   INDEX (voipCfgGkIndex)

VoipCfgGkEntry:
   SEQUENCE :
      voipCfgGkIndex—INTEGER
      voipCfgGkStatus—INTEGER
      voipCfgGkIpAddress—IpAddress)

voipCfgGkIndex OBJECT-TYPE ( voipCfgGkEntry 1)
   SYNTAX INTEGER
   ACCESS read-only
   STATUS mandatory
   DESCRIPTION This number uniquely identifies the Gatekeeper.

voipCfgGkStatus OBJECT-TYPE (voipCfgGkEntry 2)
   SYNTAX INTEGER:
      registered(1)
      not_registered(2)
   ACCESS read-only
   STATUS mandatory
   DESCRIPTION This value indicates whether the gateway is registered
   with the Gatekeeper.

voipCfgGkIpAddress OBJECT-TYPE (voipCfgGkEntry 3)
   SYNTAX IpAddress
   ACCESS read-only
   STATUS mandatory
   DESCRIPTION The IP address of the Gatekeeper.
```

```
voipCfgGwVpnMode OBJECT-TYPE (voipCfgGwGroup 1)
   SYNTAX INTEGER:
      no (1)
      yes(2)
   ACCESS read-only
   STATUS mandatory
   DESCRIPTION Virtual Private Network Toggle Switch.
voipCfgGwPktAudioMode OBJECT-TYPE (voipCfgGwGroup 2)
   SYNTAX INTEGER:
      other(1)
      g711_ulaw(2)
      g711_alaw(3)
      g723(4)
      g729(5)
      g723_6_4kps(6)
   ACCESS read-only
   STATUS mandatory
   DESCRIPTION Audio Coder to be used for voice packetization.
```

The `voipCfgGwVpnMode` and `voipCfgGwPktAudioMode` objects can be accessed using index 0 because they are separate leaves in the MIB tree.

The `voipCfgGkIndex`, `voipCfgGkCurrent` and `voipCfgGkIpAddress` objects are located in the `voipCfgGkTable` table. They can be obtained using `voipCfgGkIndex` as an index.

SNMP: Traps for VoIP-related conditions

With MAX TNT TAOS 8.0.1, VoIP-enabled MAX TNT units can generate traps for the following MultiVoice Gateway events:

- Change in the call logging server
- Change in configured Gatekeeper for VoIP
- Change in state of a WAN line

For the traps to be sent, traps must be enabled in the system and the individual trap conditions must be set to `yes`. For details about enabling traps, see the *MAX TNT Administration Guide*. Following are the relevant parameters (shown with default values) for enabling the individual trap conditions:

```
[in TRAP/""]
call-log-serv-change-enabled = no
voip-gk-change-enabled = no
wan-line-state-change-enabled = no
```

Parameter	Specifies
Call-Log-Serv-Change-Enabled	Enable/disable trap generation when the call-logging server changes. If the call-logging server index is changed or if the IP address of the active call-logging server is changed, this trap sends the following information to the SNMP manager: • The new call logging server index (callLoggingServerIndex) • The IP address of new call logging server (callLoggingServerIPAddress) • The absolute time to show when the server change occurred (sysAbsoluteCurrentTime) (Ascend Trap 38)
Voip-GK-Change-Enabled	Enable/disable trap generation when the registered Gatekeeper changes. If a new Gatekeeper is registered with the Gateway, a register request (RRQ) message is sent from the Gateway to the new Gatekeeper. When the Gateway receives the admission request (ARQ) message from the new Gatekeeper, this trap sends the following information to the SNMP manager: • The new Gatekeeper index (voipCfgGkIndex) • The IP address of new Gatekeeper (voipCfgGkIpAddress) • The absolute time to show when the Gatekeeper change occurred (sysAbsoluteCurrentTime) (Ascend Trap 39)
WAN-Line-State-Change-Enabled	Enable/disable trap generation if the state of an E1 or T1 line changes. This trap sends the following information to the SNMP manager: • The T1 or E1 line interface index (wanLineIfInde) • The line usage (wanLineUsage). This usage is reported as `trunk`, `quiesced`, or `disabled`. • The absolute time to show when the line state changed (sysAbsoluteCurrentTime) (Ascend Trap 40)

NavisAccess support for VoIP call reporting

MAX TNT TAOS 8.0.1 supports basic VoIP call reporting using NavisAccess. This includes the capability to generate Start records, Stop records, and Call Progress records for both VoIP and fax calls. These records allow NavisAccess to monitor Gateway resource usage and provide information to create billing records. Each VoIP call can generate two or more records.

Start records

A Start record reports the point in the call where a speech communications is established. Start records can provide the following information:

Attribute	Specifies
Ascend-Call-Direction	Direction of the call between the Gateway and PSTN. The reported values are Ascend-Call-Direction-Incoming (0) and Ascend-Call-Direction-Outgoing (1). (Ascend Trap 48)
NAS-Port	Encoded NAS port used for this call. (RFC Trap 5)

Attribute	Specifies
NAS-Port-Type	Encoded NAS port used for this call. The value 7 for this attribute identifies a VoIP call. (RFC Trap 61)
NAS-IP-Address	NAS IP address associated with this call. (RFC Trap 4)
Session-Id	NAS session index recorded in the session table for this call. (RFC Trap 44)
Ascend-Modem-PortNo	DSP/modem port allocated for processing this call. This value is part of the resource count information, and is repeated each time it is allocated for a call. (Ascend Trap 120)
Ascend-Modem-SlotNo	Slot where the DSP/modem card associated with the reported Ascend-Modem-PortNo is located. This value is part of the resource count information, and is repeated each time it is allocated for a call. (Ascend Trap 121)
Ascend-Modem-ShelfNo	Shelf where DSP/modem card allocated for processing this call is installed. This is part of the resource count information, and is repeated each time it is allocated for a call. (Ascend Trap 122)
Called-Station-Id (DNIS)	Dialed number string reported by the Gateway for the called destination. (RFC Trap 30)
Ascend-Dialed-Number	Dialed number string used by the Gateway to complete the call. (Ascend Trap 24)
Service-Type	Requested type of service, the value of the Type of Service byte, for this call. (RFC Trap 6)
Ascend-H323-Destination-NAS-ID	NAS IP address used to route the call to the connecting Gateway. (Ascend Trap 22)
Ascend-H323-Gatekeeper-IP	IP address of the Gatekeeper used to route the call. The Gateway is registered with this Gatekeeper. (Ascend Trap 19)
Ascend-Global-Call-Id	IP address used by the Gatekeeper to identify the connecting Gateway for this call. (Ascend Trap 20)
Ascend-H323-Conference-ID	IP address used to identify the called destination. (Ascend Trap 21)
Ascend-H323-Presession-Time	Time from the moment the caller finishes dialing the destination telephone number until the moment the speech path is established to the called destination. (Ascend Trap 198)
Ascend-H323-Dialed-Time	Time the user spends dialing the destination telephone number. This value will be zero for call originating from the LAN. (Ascend Trap 23)
Ascend-Session-Type	Audio codec used for processing the call. (Ascend Trap 18)

Stop records

A Stop record is generated at the moment when MultiVoice begins to tear down the speech path or when an incoming call to a Gateway fails to connect. A Start record can contain following information:

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Attribute	Specifies
Acct-Session-Time	Time from the moment the speech path is established to the called destination until the moment MultiVoice begins to tear down the speech path. (RFC Trap 46)
Ascend-Connect-Progress	A number that represents the call connect state at the time the call was terminated. (Ascend Trap 195)
Ascend-Disconnect-Cause	A number that reports the H.323 call disconnection reason. (Ascend Trap 196)
Ascend-H323-Inter-Arrival-Jitter	Estimated interarrival jitter for voice packets received by a Gateway. (Ascend Trap 25)
Ascend-Dropped-Octets	The number of voice frames (in bytes) dropped by a Gateway during call processing. (Ascend Trap 26)
Ascend-Dropped-Packets	Number of voice packets dropped by a Gateway during call processing. (Ascend Trap 26)
Acct-Input-Octets	Number of voice frames (in bytes) received by a Gateway during this call. (RFC Trap 42)
Acct-Input-Packets	Number of voice packets received by a Gateway during this call. (RFC Trap 47)
Acct-Output-Octets	Number of voice frames (in bytes) sent by a Gateway during this call. (RFC Trap 43)
Acct-Output-Packets	Number of voice packets sent by a Gateway during this call. (RFC Trap 48)

Call Progress records

A Call Progress record can be generated during a VoIP call when a change in resource occurs for a fax or transparent modem call. For fax calls, this record includes the modem speed and modulation. A progress message contains all the information included in a Start record.

Exhibit C-5

Lucent Technologies
Bell Labs Innovations

MultiVoice® for APX™/MAX TNT®

Configuration Guide

Part Number: 7820-0651-005
For software version 10.0
July 2002

Notice

Every effort was made to ensure that the information in this document was complete and accurate at the time of printing, but information is subject to change.

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- Type of computer you are using
- Description of the problem

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